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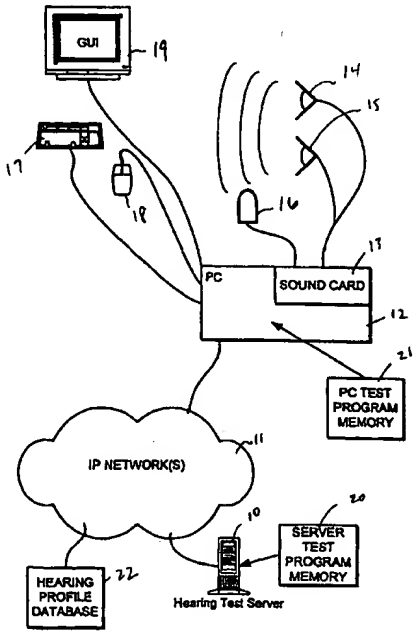
(54) Title: INTERNET BASED HEARING ASSESSMENT METHODS



(57) Abstract: A method for conducting a hearing assessment test that can be administered via the Internet using consumer quality multimedia components are provided. The method for conducting a hearing test using a computer program that comprises establishing a communication channel between an end station and a server in the communication network, such as the Internet. The method involves executing a first portion of the computer program at the server and a second portion of the computer program at the end station. The allocation of processing logic between the first portion of the computer program executed the server, and a second portion of the computer program executed in station is made according to particular implementations of the computer program, such that for example the

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first portion of the computer program includes logic for managing control of the test, and a second portion of the computer program includes logic for generating stimuli during the test, and accepting input data from the user during the test. The expertise required for successful hearing assessment is provided using resources accessible via the internet, rather than in reliance the presence of an audiologist during the test. Techniques for calibrating a sound card, such as on a home computer in support of the hearing test are included.

INTERNET BASED HEARING ASSESSMENT METHODS

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BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates the field of audiology, and more particularly to techniques for producing data indicating an assessment of an individual's hearing profile facilitated by the use of the Internet or other communication network technology.

Description of Related Art

Assessing an individual's hearing profile is important in a variety of contexts. For example, individuals with hearing profiles that are outside of a normal range, must have their profile recorded for the purposes of prescribing hearing aids which fit the individual profile. Typically hearing profile assessments are made by professional audiologists using calibrated and specialized equipment. Therefore, hearing profile assessments are relatively difficult to obtain and expensive.

Because of the difficulty in obtaining a hearing assessment test, and for a variety of other reasons, many persons who could benefit from devices that would assist their hearing do not follow through with obtaining a prescription for such devices. Thus, it is desirable to simplify the procedures involved in obtaining a reliable hearing assessment.

United States Patent No. 5,928,160 describes a home hearing test system and method based on the use of calibrated headphones specially manufactured to support the hearing test using home audio equipment. In addition, reference is made to this patent for its discussion of background concerning hearing assessment tests in general. However, home hearing assessment tests have not achieved commercial acceptance. Professional audiologists are typically required because of the large number of factors involved in making an assessment necessary for generating a reliable hearing profile. An audiologist is able to set up a controlled environment, and conduct the test according to a testing protocol involving a number of stimuli and response

steps that is adapted based on the responses gathered during the test.

A variety of uses for hearing profiles, other than for the purposes of prescribing hearing aids and assistive listening devices, is being developed. For example, hearing profiles of individuals can be utilized for producing customized audio products, such as pre-recorded music that has been modified according to the hearing profile of the listener. One medium for delivering customized audio products is the Internet. See, commonly owned and copending U.S. Patent Application No.: 09/464,036, filed 15 December 1999, by Pluvinage, et al., entitled "System and Method for Producing and Storing Hearing Profiles and Customized Audio Data Based on Such Hearing Profiles."

As the Internet gains popularity, and more individuals obtain the general-purpose processing power of personal computers coupled to the Internet and having sound cards or other audio processing capability, the Internet is becoming a more important medium for the delivery of audio products. Accordingly, it is desirable to leverage the communication technology the Internet used in the delivery of audio products for the purposes of performing hearing assessments in the home.

SUMMARY OF THE INVENTION

The present invention provides methods and systems for conducting a hearing assessment test that can be administered via the Internet using consumer quality multimedia components. Many ways to implement such a test are described. Aspects and advantages of the present invention include:

1) The expertise associated with following a test protocol is delivered to consumer via the internet.
2) The control functions used for following a test protocol implemented by internet based program.
3) The test protocol may be structured like a standard audiometric test or it may make full use of the mutli-media capabilities of the computer, or other processor, and embed the protocol within an audiovisual experience (e.g. game, task set).
4) A computer sound card or other audio processor, as found in home computers coupled to the internet, is used for sound generation and for sound measurements.

5) The test protocol may be used for assessing hearing loss across audible frequency range or any subset of the audible range.

6) The test is initiated by a user, and run at the user's internet connected computer, hand held device or cell phone. However, the expert administration of the test needed to ensure valid results are imbedded in the test software delivered via the internet.

The present invention provides a method for conducting a hearing test using a computer program that comprises establishing a communication channel between an end station and a server in the communication network, such as the Internet. The method involves executing a first portion of the computer program at the server and a second portion of the computer program at the end station. The allocation of processing logic between the first portion of the computer program executed the server, and a second portion of the computer program executed at the end station is made according to particular implementations of the computer program, such that for example the first portion of the computer program includes logic for managing control of the test, and a second portion of the computer program includes logic for generating stimuli during the test, and accepting input data from the user during the test. Processing of the input data is executed either locally on the user's computer, at the server, or in both places. In one aspect of the invention, the computer program used for executing the test includes a component to deliver the second portion of the computer program to the end station from a resource coupled to the communication network, such from the server itself, or from a database accessible via the communication network. In this case the second component executed at the end station may include any combination of the logic to control the test, to process the data gathered in the test, and to produce the sound signals used in the test.

The computer program in various aspects of the invention produces audio stimuli during the test, except input data from a user according to a testing protocol. In addition, environmental data is sensed at the user's end station during the test, for use in assessing background noise or other control factors processed in combination with the input data to produce a hearing assessment. In one embodiment, the process includes calibrating the sound card or other sound processing resources on the end station.

There are a number of different test types that can be used within an Internet based

hearing loss assessment utilizing first and second portions of a computer program as described above. The different types are dependent on the type of data. A variety of suitable test types includes:

Hearing threshold level

Masking threshold level

Loudness matching

Loudness Growth in Octave Bands (LGOB)

Speech Reception Threshold and Speech Discrimination in noise or quiet

Temporal Compression/Expansion/Masking

According to one aspect of the invention, the component of the test program which is executed at the end station includes a graphical user interface which supports a multimedia presentation of the test. The constructs used on the graphical user interface for prompting input data from a user during the test depend on the particular test protocol being executed. The character of the graphical user interface in various embodiments approximates a game or other task set organization.

The present invention is applied on end stations capable of internet communications, and including audio capability, including home computers, laptop computers, hand held or palm sized computers, internet-enabled mobile phones, and the like.

According to yet another aspect of the invention, the computer program used for executing the hearing test includes logic that manages a calibration routine for the sound card or other audio resources coupled to the end station of the user, and used for producing the audio stimuli in the test. Alternatively, or in addition, the computer program used for executing the hearing test includes logic that accounts for background noise, attributes of the audio resources coupled to the end station of the user, or other environmental factors detected during the test.

According to another aspect of the invention a calibration device is provided that is suitable for use in electronic calibration of sound cards on end stations for supporting the test program of the present invention. The calibration device may also be used for other purposes requiring calibrated audio signal generation from sound cards having variant architectures. The calibration device includes a test signal source and a switch. The switch is operated in one embodiment in response to DTMF or other format control signals produced by the sound card to be tested. The switch is operated so that test signal source is used to determine one of the input

and output transfer functions of the sound card under test, and the other of the transfer functions is determined using feed back.

Other aspects and advantages of the present invention can be seen upon review the figures, the detailed description and claims which follow.

BRIEF DESCRIPTION OF THE DRAWING

Fig. 1 illustrates an Internet based system for conducting a hearing assessment test according to the present invention.

Fig. 2 is a flow chart illustrating the method of operation for an Internet based test according to present invention.

Fig. 3 is a block diagram of a test configuration for actual sound pressure level measurement inferred from drive voltage for use according to the present invention.

Fig. 4 is a block diagram of a test configuration for actual sound pressure level measurement using a voltage controlled oscillator VCO.

Fig. 5 is a block diagram of a test configuration for actual sound pressure level measurement completed at the ear using a calibration source box and a calibrated microphone.

Fig. 6 is a block diagram of a test configuration for actual sound pressure level measurement completed at the ear using a VCO mapping function.

Fig. 7 is a block diagram of a test configuration for relative sound pressure level measurement.

Fig. 8 is a graph showing average audiometric data by age group, representing typical audiograms which may be produced using the present invention.

DETAILED DESCRIPTION

A detailed description of preferred embodiments of the present invention is provided with respect to Fig. 1 through Fig. 8, in which Figs.1 and 2 provide illustrate a system and method for Internet based hearing assessment tests. Figs. 3 through 8 provide description of various testing configurations at user end stations suitable for use with present invention.

Fig. 1 illustrates the Internet based system of the present invention implementing a hearing assessment test. System includes a hearing test server 10 coupled to a communication network 11, such as the Internet. A user end station 12, such as a personal computer, is also

coupled to the communication network 11. The end station 12 includes a soundcard 13 which provides data processing resources for producing audio output and receiving audio input under control of the logic in computer programs executed by the processor in the end station 12. In the figure, the sound card 13 is connected to stereo speakers 14 and 15, or more preferably to a headphone, and to a microphone 16. The end station 12 also includes a display 19, a keyboard 17, and a mouse 18. During the test, audio stimuli in the form of sound signals produced in the soundcard 13 are generated using the stereo speakers 14 and 15. The sound signals may be sampled or computed sound. Environmental factors such as background noise, and the level of the output of the speakers 14 and 15 is sensed using a microphone 16. The display 19 is used to display a graphical user interface which prompts a user to input data using the keyboard 17 or the mouse 18 in response to the audio stimuli of the test.

Headphones used in the preferred embodiment include a microphone inside used for calibration and sensing of the environment during the test. Also, the headphones include seating arrangement to filter noise from the background.

The hearing test is executed using a computer program that includes a first component stored on the server test program memory 20 which is connected to the server 10, and a second component which is stored in the PC test program memory 21 which is connected to the end station 12. Upon completion of a test, a hearing profile is produced for the user. In a preferred system, this hearing profile is stored in a hearing profile database 22 which is accessible using Internet 11. In another embodiment, the hearing profile database 22 is coupled directly to the server 10. Alternatively, the hearing profile might be stored only on users end station and not made available to the communication network.

In this example, the end station 12 consists of a personal computer with standard soundcard components. In various embodiments, additional components may be added, including a microphone preamplifier to increase microphone output signal levels, a soundcard output accumulator for managing the output levels for the audio stimuli, and headphones with or without built-in microphones.

In one implementation, the hearing test server 10 maintains a web site. To initiate a hearing test, a user at the end station 12 accesses the web site and downloads a component of the hearing test computer program from the server 10 for execution at the end station 12. The user initiates the test without intervention by a third party, and uses the resources available via the

Internet and the resources at the end station to conduct a hearing test.

Fig. 2 illustrates the basic flowchart for the process of performing an Internet based hearing assessment test. In a first step 50, a user establishes a link between the end station and a hearing test server via a communication network such as the Internet. In one example, the link comprises a connection according to the transmission control protocol executing over the Internet protocol TCP/IP. The link may also involve protocols like the hypertext throughput protocol HTTP, and other Internet protocols.

In a next step 51, test control resources and data processing resources that will be utilized during the test are allocated. The allocation of these resources can take a variety of configurations, including maintaining all of the resources at the server, and providing an Internet based interface accessible using a browser or email client at the end station, maintaining the test control resources at the Internet server, and data processing resources at the end station, or other combinations as suits the particular implementation of the program to control the test and to process the data generated during the test.

In step 52, test sound signal resources are allocated. The sound signal resources may include sound samples, programs for generating sounds, or other common sound synthesis tools. The sound signal resources are adapted to the particular type of hearing test to be executed. In one embodiment, the sound signal resources are downloaded to the end station from the server. In another embodiment, the sound signal resources are available in the personal computer soundcard without requiring download from the server, such as by providing recorded audio files with drivers for sound cards that are loaded on a user's end station. In another embodiment, sound signal resources are distributed between the end station and a server during execution of the test.

Next, calibration programs are executed to evaluate the test environment (the audio environment in which the end station is situated), and the test set up (the audio characteristics of the equipment at the end station (step 53).

Upon completion of the allocation of data processing resources and calibration, test control resources and sound signal resources necessary for supporting the test, the test is initiated. The first step in the test is to generate a sound using the sound signal resource (block 54). Next, the process accepts and processes input using the test control and test data processing resources (block 55). Next the routine determines whether the test has been completed (block

56). If the test is not completed, then the algorithm determines a next sound according to a test protocol being executed using the test control resources, in response to the input from the user and the state of the test in one implementation (block 57). Then, the process loops back to step 54 to generate the next sound. If at block 56, it is determined at the test is completed, then the hearing profile is stored (block 58).

There are numerous options for prompting feedback from a test subjects. Options include accepting input in the form of the keystroke, a mouse click, use of a selection button or a timeout interval as volume is adjusted by the test control resources, or by the test subject action of increasing the volume, until some criterion is reached. A second option for accepting input includes causing the user to complete in action prompted by the graphical user interface when test generated sounds meet some criterion. For example, the test sound may be a sound varying in loudness. The test subject enters a mouse click when the sound disappears.

The test control resources can be distributed between the server and the end station using an Internet link, or using an executable file downloaded from the server and run locally on the test subjects equipment, or any partitioning of control in between. By controlling the test flow, the program can provide expertise for measuring and evaluating the level of background noise, testing for false positives and false negatives, and in general control the flow and pace of the test process according to a test protocol. Controlling the test flow has specific advantages toward maintaining test subject interest as the user can be prompted to provide appropriate feedback and responses.

In other embodiments, data collected during the test can be returned to the web site server as raw data, as completely analyzed result data such as a hearing profile, or as any combination of raw and processed data in between. In one embodiment, data is not returned web site server in all, but rather completely processed locally on the end station using resources downloaded partially or completely from the test server.

The sound signals used in the testing process are implement in several alternative forms. The type of test signals used can have significant influence on the results of the test through a number of psychoacoustic effects. A large number of possible test signals are applicable to any of the implementations. Examples of the types of test tones claimed are:

- Pure tones of long duration and constant intensity in each test step utilizing a number of different test steps at different frequencies.

- Pulses of pure tones and constant intensity in each step utilizing a number of different test steps at different frequencies.
- Combination of tones of long duration and varying intensity in each test step utilizing a number of different test steps at different frequencies.
- Pulses of combinations of tones and varying intensity in each step utilizing a number of different test steps at different frequencies.
- Constant amplitude, swept frequency sound in each test step utilizing different test steps at different amplitudes.
- Constant amplitude pulses of swept frequency sound in each test step utilizing different test steps at different amplitudes.
- Bandpass filtered noise combined with test signals.
- Speech sound with and without noise background with or without temporal compression or elongation.

Furthermore the method of test sound signal generation is not limited, and can include sampling using standard formats like MIDI, FM synthesis, wavetable synthesis or other sound generation techniques.

As mentioned before, a wide variety of hearing test protocols can be utilized for producing a hearing assessment. The particular test chosen depends on a variety of factors, including the use to which the hearing profile will be put, the type of equipment used at the end station, and any information about the physiology of the test subject which may affect the choice of hearing test. Example test type include:

1) Hearing threshold level.

The hearing threshold level test is related to identifying the sound level when the test subject can just begin to hear the test signal. This test type may be associated with determining that actual sound pressure level SPL of thresholds across the frequency range or the test method be simply to establish the relative level of thresholds as a function of frequency.

2) Masking threshold level.

The masking threshold level identifies the test signal sound level when the test signal can be heard out of a masking signal. The masking threshold test protocol can be completed at a number of different baseline amplitudes to give an indication of recruitment. This method may have some advantages when there is some background noise at frequencies other than the test

frequency.

3) Loudness matching.

In a loudness matching method, the generated sound consists of two different frequencies. One frequency is considered a baseline and is constant throughout a test. The other sound, the test sound, has a variable frequency during the test. A measurement consists of determining the loudness of the test sound that matches loudness of the baseline sound as a function of frequency. The resulting measurements are used to generate an equal loudness curve. The difference between the equal loudness curve obtained here and the equal loudness curve for normal hearing populations gives the hearing loss assessment. This test protocol can be completed at a number of different baseline amplitudes to give an indication of recruitment.

4) Loudness Growth in Octave Bands (LGOB).

Loudness Growth in Octave Bands is a subjective loudness evaluation procedure in which the test subject a prescribed set of common adjectives (e.g.very quiet, quiet, comfortable, loud, very loud and uncomfortably loud) to "measure" the loudness of test signals. The difference between the perceived loudness reported by the subject and a population of normally hearing individuals gives a measure of recruitment.

5) Speech Reception Threshold and Speech Discrimination in Noise or Quiet.

These test methods are based on the fact that different speech sounds have different frequency spectra and so the speech reception/discrimination capabilities of a subject are dependent on the subject hearing profile. Furthermore, noise can be used to test the breadth of the auditory filter. Test with background noise are particularly interesting for internet administered test because the controlled noise level can be set to mask the environmental noise.

6) Temporal Compression/Expansion/Masking.

Temporal compression of speech signals or tones can be used to probe auditory capabilities since it is known that temporal masking is affected by sensineural hearing impairment.

The test methods outlined above could be implemented in either a monaural or a binaural configuration. In the monaural implementation, each ear is tested individually and the other ear is "plugged" or otherwise deprived of test signal input. In an implementation scheme in which the headphones are supplied, the supplied headphones may have only one speaker. Clearly, there are advantages and disadvantages associated with either test method implementation with

respect to accuracy and test complexity.

The basic test methods outlined above can be implemented within a number of different test configurations. The different test configurations may have different peripheral equipment, test protocols and they may have different levels of accuracy.

Four general test configurations are based on the four possible independent combinations of the use of one of actual sound pressure level SPLa measurement and relative sound pressure level SPLr measurement, and the use of one of sensing of SPLa or SPLr at the ear and inferring SPLa or SPLr from drive voltages. One can either measure an absolute pressure or a relative pressure and that the measurement can be either an actual pressure measurement as with a microphone or it can be a measurement that infers pressure (ie measure the voltage and know the voltage to pressure transfer function). So one can make any one, any combination of more than one, or all, of the following 4 unique

measurements:

1) relative measurement using actual pressure measurements,

2) relative measurements using an inferred measurement (voltage),

3) absolute measurement using actual pressure measurements,

4) absolute measurement using an inferred measurement (voltage).

Actual SPL measurement methods measure the actual sound pressure level SPLa that coincides with the test criterion. As such, these measurements require measurement of a quantity that can be related to sound pressure level.

The SPLa can be inferred from drive voltages, rather than from audio sensors like microphones. This general method refers to the approach of using a direct measurement of the sound sources drive voltage and knowledge of the sound source's voltage-to-sound transfer function to estimate the SPL delivered during the test. Since the actual sound pressure level delivered to the ear from a given source depends greatly on the relative geometry of the source and the ear, this test method is probably requires that earphones generate the sound.

In the configuration shown in Fig. 3, a calibration source box is used in conjunction with calibrated earphones for measurement of SPLa based on inference from drive voltages. The test configuration shown in Fig. 3 includes the test server implemented as a WebSite 100. A communication link 101 is established between the WebSite 100 and the remote computer 102 acting as an end station for the test. Test control commands and information input and output

are provided across the link 101 between the WebSite 100 and remote computer 102. The remote computer 102 includes input control 103 for accepting test subject input and a sound card 104 for producing audio stimuli and accepting audio input from sensors. A calibration source box 105 is coupled to the sound card 104. The source box includes a first switch 106 and a second switch 107. One output of the calibration source box 105 is calibrated headphones 106 for use by the test subject.

The calibration source box 105 is used to determine the specific input and output transfer functions of the sound card input A/D and output D/A conversions. Once the input A/D and output D/A calibrations are known, it is possible to accurately know the output drive signal. Combining this calibration knowledge with the known headphone calibration, the absolute sound pressure level can be estimated. The calibration source box 105 is a three port box containing two switches 106, 107 and three switch contacts A, B, C. It may be configured with its own power source through either a battery connection or through a wall plug.

The calibration and test process is as follows:

Step number	Switch 106	Switch 107	SoundCard Output	SoundCard Input	Measurement
1	A or B	C	No output	Calibration Source	Determines the transfer function of the input A/D.
2	B	B	Calibration Signal Suite	Test Signal	Using the results from step 1, determines the transfer function of the output A/D.
3 repeat @ all freq. values	A	B	Test Signal Suite	No input	Test subject signals when a test criteria is met. Using data from step 1, the microphones calibration data and the microphone output, the corresponding SPL can be determined.

A simulated earphone load may need to be attached to connection point B during step 2 of the test to ensure that the output impedance seen by the sound card during calibration is the same as the load seen during the test. If this extra load is needed, then another switch and terminal will be needed.

The measured sound pressure levels that meet the test criteria at the various test frequencies allows the calculation of hearing loss.

The measured sound pressure levels are the sound pressure levels that meet the test criteria at the various test frequencies. These data are then used to calculate hearing loss according to the test type.

Typically a three step calibration process is used. In the first step (Cal 1), the Signal Source is used to input a known signal into the input port of the soundcard. Measurement of the resulting digital word gives the input voltage-to-digital transfer function. This measurement is completed with the soundcard's microphone input gain set at ~1/4 of its full range. Utilizing the pressure-to-voltage transfer function of the calibrated microphone allows the pressure-to-digital transfer function of the microphone/pre-amp/sound card input path to be fully defined.

The second step (Cal 2) addresses the fact that the voltage in-to-digital conversion of sound cards is not linear over the entire range of microphone gain. Furthermore, it is known that SNR is improved if the microphone gain is maximized. Cal 2 gathers the information needed to determine what input/output gain and digital word to use to optimize signal integrity and test flow speed. To complete this characterization, the sound card input is driven with a number of different soundcard output levels at number of different microphone gain levels. The combined information gathered in Cal 1 and Cal 2 completely calibrates the microphone/microphone input portion of the system.

The third calibration step (Cal 3) is used to generate the Sound card output word to ear level sound pressure transfer function. This transfer function is determined by driving the headphones with a soundcard digital word and measuring the pressure with the previously calibrated microphone/microphone input.

In one embodiment, an active audio channel during the test delivers sound to one ear, while the other audio channel is used to deliver control signals to the calibration box, for controlling the switches for example. One suitable control signal format is the DTMF tone standard used in telephone systems.

Note that if it is assumed that the sound card's input A/D transfer function is one, then an actual SPL measurement can be made using the calibrated headphones but without the calibration source box. In the this case, calibration is not required and the headphone output can be directly fed into the input of the sound card.

Fig. 4 shows a configuration suitable for SPLa measurements using a voltage controlled

oscillator VCO based method. In Fig. 4, the configuration components that are similar to those of Fig. 3 are given like reference numbers and not described again. In this case, the calibration source box 105 of Fig. 3 is replaced by VCO 110 which receives an input voltage on line 111 and produces and output on line 112 based upon the signals being transmitted to calibrated headphones on line 113.

This method assumes that the sound card input port's frequency–to-frequency transfer function is one. Using this assumption, the VCO 110 is used to map the calibrated headphone drive voltage amplitude into frequency space. The drive voltage is determined through frequency space analysis of the VCO output combined with known headphone calibration.

In this test method, the test subject signals when the test criteria is met. Using the measured frequency of the VCO output, the known VCO mapping function and the known headphone calibration, the corresponding SPL can be determined. The measured sound pressure levels at the various test frequencies give the hearing loss assessment.

Actual sound pressure level SPLa may also be sensed at the ear using configurations like those shown in Figs. 5 and 6. This general method uses direct measurement of the sound pressure level at the ear. Here a calibrated microphone is used to measure the sound pressure level during the test and during the calibration program.

In the configuration of Fig. 5, a calibration source box 120 is used in conjunction with a calibrated microphone 121. In Fig. 5, the configuration components that are similar to those of Fig. 3 are given like reference numbers and not described again.

The calibration source 120 is used to determine the specific transfer functions of the sound card input A/D conversion. Combining this calibration knowledge with the known headphone calibration, the sound pressure level can be determined. The calibration source box 120 is a five port box containing three switches 122, 123, 124 and three switch contacts A, B, C. It may be configured with its own power source through either a battery connection or through a wall plug. The calibration and test process are shown in the following table.

Step number	Switch 122	Switch 123	Switch 124	SoundCard Output	SoundCard Input	Measurement
1	A	C	B	No output	Calibration Source	Determines the transfer function of the input A/D.
2 repeat	A	B	C	Test Signal	SPL microphone	Test subject signals when a test criteria is met. Using

@ all freq. values				Suite		data from step 1, the microphone calibration data and microphone output, the corresponding SPL can be determined.

The measured sound pressure levels are the sound pressure levels that meet the test criteria at the various test frequencies. These data are then used to calculate hearing loss according the test type.

Fig. 6 shows a configuration suitable for SPLa measurements at the ear using a voltage controlled oscillator VCO based method. In Fig. 6, the configuration components that are similar to those of Fig. 3 are given like reference numbers and not described again. In this case, the calibration source box 1120 of Fig. 5 is replaced by VCO 150 which receives an input voltage on line 151 and produces an output on line 152 based upon the signals received on line 153 from transducers 155. The transducers 155 comprise headphones or speakers coupled to the sound card 104 by line 156. The headphones embodiment includes an embedded microphone, and the speaker embodiment includes a stand alone microphone.

This method assumes that the sound card input port's frequency–to-frequency transfer function is one. Using this assumption, a VCO is used to map the calibrated microphone output voltage into frequency space. The microphone output is determined through frequency space analysis of the VCO output combined with the known microphone calibration.

In this test method, the test subject signals when the test criteria is met. Using the measured frequency of the VCO output, the known VCO mapping function and the known microphone calibration, the corresponding SPL can be determined. These data are then used to calculate hearing loss according the test type.

Relative measurement of sound pressure levels may also be used for some embodiments, as mentioned above. The SPL corresponding to a test criteria may not be as important as knowledge of the relative threshold of hearing across the audible frequency range. The following section outlines relative SPL level tests. Each method described could be implemented with either headphones or speakers. These test methods assume that sound card's input A/D and its output D/A converters are linear across frequency and amplitude. Furthermore, the transfer functions of any headphones, microphones or speakers used in the test are also assumed to be substantially linear

across frequency and amplitude. These test methods may include input pre-amps and output attenuators.

In general, the test set-ups and protocols are simpler in a relative measurement. All the test methods discussed below can use the test set-up shown in Fig. 7. In Fig. 7, the configuration components that are similar to those of Fig. 3 are given like reference numbers and not described again. In the configuration of Fig. 7, the sound card output on line 161 is supplied to a transducer/sensor 160 comprising for examples, headphones with an embedded microphone or speakers with a stand alone microphone. The microphone output is supplied on line 162 to the sound card 104.

Relative SPL measurement inferred from drive voltage is the simplest test method. The test proceeds by playing the test sounds either through the speakers or through headphones. The test subject signals when the test criterion is met. The measured data is the sound card's digital word is the output.

A method based on relative SPL measurement completed at ear requires that a microphone be used to measure the actual sound pressure level at the ear. The test configuration is shown in Fig. 7 as well. The measured data is the sound card's input digital word.

The goal of the Home Audio Test is to provide information related to the auditory capability of the subject to evaluate the subject's suitability for personalized audio. The test does not need to return any specific information about the hearing loss of the subject.

In some embodiments, it is understood that an audiogram will not provide enough information to ensure a good fit in all cases. Hence, iterations will be needed to find the correct fit. With this in mind, the goal of the test is to provide sufficient information to ensure that the first clip of reprocessed audio shows some benefit to the subject. Clearly, the first reprocessed audio can not lead to displeasing sound quality. These rather vague requirements will be translated into specific statements about accuracy, resolution, dynamic range, and frequency below.

One core of the audio reprocessing algorithm involves developing corrections for recruitment. As a result, the benefits of the algorithm will be most observed by subject with predominately semi-neural hearing loss. While the details of any individuals audiogram will show a wide range in structure, for the development of this test, using the test goal stated above and the understanding that some iterations will be needed during the fit, a prototypic hearing loss profile will be assumed. The assumed hearing loss profile will be defined by the following conditions:

- Hearing Loss profile will follow the following equations: hl=b*exp(m*freq) where m and b are constants(slope and ln(intercept on a ln(HL) vs freq plot).
- The hearing loss between 125hz and 500hz is constant and equal to the value given for 500hz from the equation given above.

The assumed shape for audiograms is shown in Fig. 8.

While the present invention is disclosed by reference to the preferred embodiments and examples detailed above, it is to be understood that these examples are intended in an illustrative rather than in a limiting sense. It is contemplated that modifications and combinations will readily occur to those skilled in the art, which modifications and combinations will be within the spirit of the invention and the scope of the appended claims.

CLAIMS

What is claimed is:

1. A method for conducting a hearing test using a computer program, comprising:

establishing a communication channel between an end station and a server in a communication network;

executing a first portion of the computer program at the server;

executing a second portion of the computer program at the end station.

2. The method of claim 1, wherein the communication network comprises a packet switched network.

3. The method of claim 1, wherein the communication network comprises a network executing according a standard internet protocol.

4. The method of claim 1, wherein the channel comprises a connection according to a standard transmission control protocol over a standard internet protocol (TCP/IP).

5. The method of claim 1, wherein the second portion of the computer program includes logic that presents a set of stimuli to a user at the end station, and accepts input from the user responsive to the stimuli.

6. The method of claim 1, wherein the second portion of the computer program includes logic controlling a sensor at the end station to sense environmental data at the end station during the test.

7. The method of claim 1, wherein the second portion of the computer program includes logic controlling a sensor at the end station to sense a set up at the end station during the test.

8. The method of claim 1, wherein the computer program includes test control, test data processing, and test sound signal components which are distributed between said first and second portions.

9. The method of claim 1, wherein the computer program includes a component to deliver the second portion of the computer program to the end station from a resource coupled to the communication network.

10. The method of claim 9, wherein the resource comprises memory at the server.

11. The method of claim 8, wherein the test control component executes a protocol responsive to the test data processing component involving interaction according to the input from the user and the environmental data sensed during the test.

12. The method of claim 8, wherein the test control component executes a protocol responsive to the test data processing component involving interaction according to the input from the user and the test set up data sensed during the test.

13. The method of claim 5, wherein the second component includes logic driving a graphical user interface to the user in conjunction with the set of stimuli, prompting the user to provide said input.

14. The method of claim 1, wherein the first portion manages presentation of stimuli for the test and the second portion controls production of the stimuli at the end station.

15. The method of claim 1, wherein the hearing test comprises a hearing threshold level test.

16. The method of claim 1, wherein the hearing test comprises a masking threshold level test.

17. The method of claim 1, wherein the hearing test comprises a loudness matching test.

18. The method of claim 1, wherein the hearing test comprises a loudness growth in octave bands LGOB test.

19. The method of claim 1, wherein the hearing test comprises a speech reception threshold and

speech discrimination in noise and quiet test.

20. The method of claim 1, wherein the hearing test comprises a temporal compression/expansion/masking test.

21. The method of claim 1, wherein the computer program comprises logic to make measurements to determine actual sound pressure levels of thresholds involved in the test.

22. The method of claim 1, wherein the computer program comprises logic to make measurements to determine relative sound pressure levels of thresholds involved in the test.

23. The method of claim 1, wherein the end station comprises an internet enabled mobile phone.

24. The method of claim 1, wherein the end station comprises a home computer.

25. The method of claim 1, wherein the end station comprises a hand held computing platform

26. A method for conducting a hearing test using a computer program, comprising:

linking an user end station to a server using a communication network;

allocating test control and data processing resources between the user end station and the server;

allocating test sound signal resources to the user end station;

generating a sound using the test sound signal resource;

accepting and processing input using the test control and data processing resources;

determining a status of a test according to a test protocol, and if the test is done, then storing a hearing profile for the user, and if the test is not done, then determining a next stimulus according to the test protocol using the test control resources, and returning to the step of generating a sound.

27. The method of claim 26, wherein the communication network comprises a packet switched network.

28. The method of claim 26, wherein the communication network comprises a network executing according a standard internet protocol.

29. The method of claim 26, wherein the channel comprises a connection according to a standard transmission control protocol over a standard internet protocol (TCP/IP).

30. The method of claim 26, wherein including logic controlling a sensor at the end station to sense environmental data at the end station during the test.

31. The method of claim 30, wherein the test control resources execute a test protocol responsive to the test data processing resources involving interaction according to input from the user and the environmental data sensed during the test.

32. The method of claim 30, wherein the set of stimuli includes audio stimuli, and the environmental data comprises background noise.

33. The method of claim 26, wherein including logic controlling a sensor at the end station to sense test set up data at the end station during the test.

34. The method of claim 33, wherein the test control resources execute a test protocol responsive to the test data processing resources involving interaction according to input from the user and the test set up data sensed during the test.

35. The method of claim 26, including logic driving a graphical user interface in conjunction with generating the sound, which prompts the user to provide input.

36. The method of claim 26, wherein the end station comprises an internet enabled mobile phone.

37. The method of claim 26, wherein the end station comprises a home computer.

38. The method of claim 26, wherein the end station comprises a hand held computing platform.

39. A method for conducting a hearing test using a computer program, comprising:

establishing a communication channel between an end station and a server in a communication network;

executing a first portion of the computer program at the server;

executing a second portion of the computer program at the end station, wherein the end station includes sound processing resources for producing audio signals during the test; and

calibrating the sound processing resources.

40. The method of claim 39, wherein the communication network comprises a packet switched network.

41. The method of claim 39, wherein the communication network comprises a network executing according a standard internet protocol.

42. The method of claim 39, wherein the channel comprises a connection according to a standard transmission control protocol over a standard internet protocol (TCP/IP).

43. The method of claim 39, wherein the second portion of the computer program includes logic that presents a set of stimuli to a user at the end station, and accepts input from the user responsive to the stimuli.

44. The method of claim 39, wherein the second portion of the computer program includes logic controlling a sensor at the end station to sense environmental data at the end station during the test.

45. The method of claim 39, wherein the second portion of the computer program includes logic controlling a sensor at the end station to sense a set up at the end station during the test.

46. The method of claim 39, wherein the computer program includes test control, test data processing, and test sound signal components which are distributed between said first and second

portions.

47. The method of claim 39, wherein the computer program includes a component to deliver the second portion of the computer program to the end station from a resource coupled to the communication network.

48. The method of claim 47, wherein the resource comprises memory at the server.

49. The method of claim 46, wherein the test control component executes a protocol responsive to the test data processing component involving interaction according to the input from the user and the environmental data sensed during the test.

50. The method of claim 46, wherein the test control component executes a protocol responsive to the test data processing component involving interaction according to the input from the user and the test set up data sensed during the test.

51. The method of claim 43, wherein the second component includes logic driving a graphical user interface to the user in conjunction with the set of stimuli, prompting the user to provide said input.

52. The method of claim 39, wherein the first portion manages presentation of stimuli for the test and the second portion controls production of the stimuli at the end station.

53. The method of claim 39, wherein the hearing test comprises a hearing threshold level test.

54. The method of claim 39, wherein the hearing test comprises a masking threshold level test.

55. The method of claim 39, wherein the hearing test comprises a loudness matching test.

56. The method of claim 39, wherein the hearing test comprises a loudness growth in octave bands LGOB test.

57. The method of claim 39, wherein the hearing test comprises a speech reception threshold and speech discrimination in noise and quiet test.

58. The method of claim 39, wherein the hearing test comprises a temporal compression/expansion/masking test.

59. The method of claim 39, wherein the computer program comprises logic to make measurements to determine actual sound pressure levels of thresholds involved in the test.

60. The method of claim 39, wherein the computer program comprises logic to make measurements to determine relative sound pressure levels of thresholds involved in the test.

61. The method of claim 39, wherein the end station comprises an internet enabled mobile phone.

62. The method of claim 39, wherein the end station comprises a home computer.

63. The method of claim 39, wherein the end station comprises a hand held computing platform

64. The method of claim 39, wherein said calibrating includes:

determining an input transfer function and an output transfer function for the sound processing resources.

65. The method of claim 39, wherein said calibrating includes:

electronically determining an input transfer function and an output transfer function for the sound processing resources.

66. The method of claim 39, wherein said sound processing resources have an electronic input adapted to receive analog voltage inputs representative of sound, and first and second electronic outputs adapted to supply analog voltages representative of sound, and wherein said calibrating includes:

coupling a calibration device to the electronic input and the first and second electronic outputs of the sound processing resources; and

using the calibration device to supply a test signal to the electronic input, and feeding back a processed signal output on one of the first and second electronic outputs to the electronic input.

67. The method of claim 66, including supplying control signals to the calibration device using the other of the first and second electronic outputs.

68. The method of claim 67, wherein the control signals comprise dual tone multi-frequency DTMF signals.

69. The method of claim 66, wherein the test signal comprises a tone, and including determining an input transfer function in response to the tone, and then generating the processed signal using the sound processing resources and determining an output transfer function in response to the processed signal and the input transfer function.

70. The method of claim 66, wherein the test signal comprises an output of a voltage controlled oscillator, and including using a signal from the sound processing resources to control the voltage controlled oscillator, and determining an output transfer function in reponse to the test signal, and determining an input transfer function in response to the processed signal and the output transfer function.

71. An apparatus for calibrating sound processing resources on an end station using a program executed by the an end station, comprising:

a test signal source;

a first input adapted to receive electronic inputs representative of sounds from a first output of the sound processing resources;

a second input adapted to receive electronic inputs representative of sounds from a second output of the sound processing resources;

an output adapted to provide electronic outputs representative of sounds to a first input of the sound processing resources; and

a switch to connect the test signal source to the output, and to connect one of the first and second inputs to the output in response to control signals.

72. The apparatus of claim 71, wherein the test signal source comprises a tone generator adapted for connection to the output for use in measuring an input transfer function of the sound processing resources.

73. The apparatus of claim 71, wherein the test signal source comprises a voltage controlled oscillator, having a control input adapted to be connected to one of the first and second inputs, and to supply a signal to the output for use in measuring an output transfer function of the sound processing resources.

74. The apparatus of claim 71, including a circuit supplying signals from one of the first and second inputs as the control signals for the switch.

75. The apparatus of claim 74, wherein the signals from the one of the first and second inputs comprise dual tone multi-frequency DTMF signals.

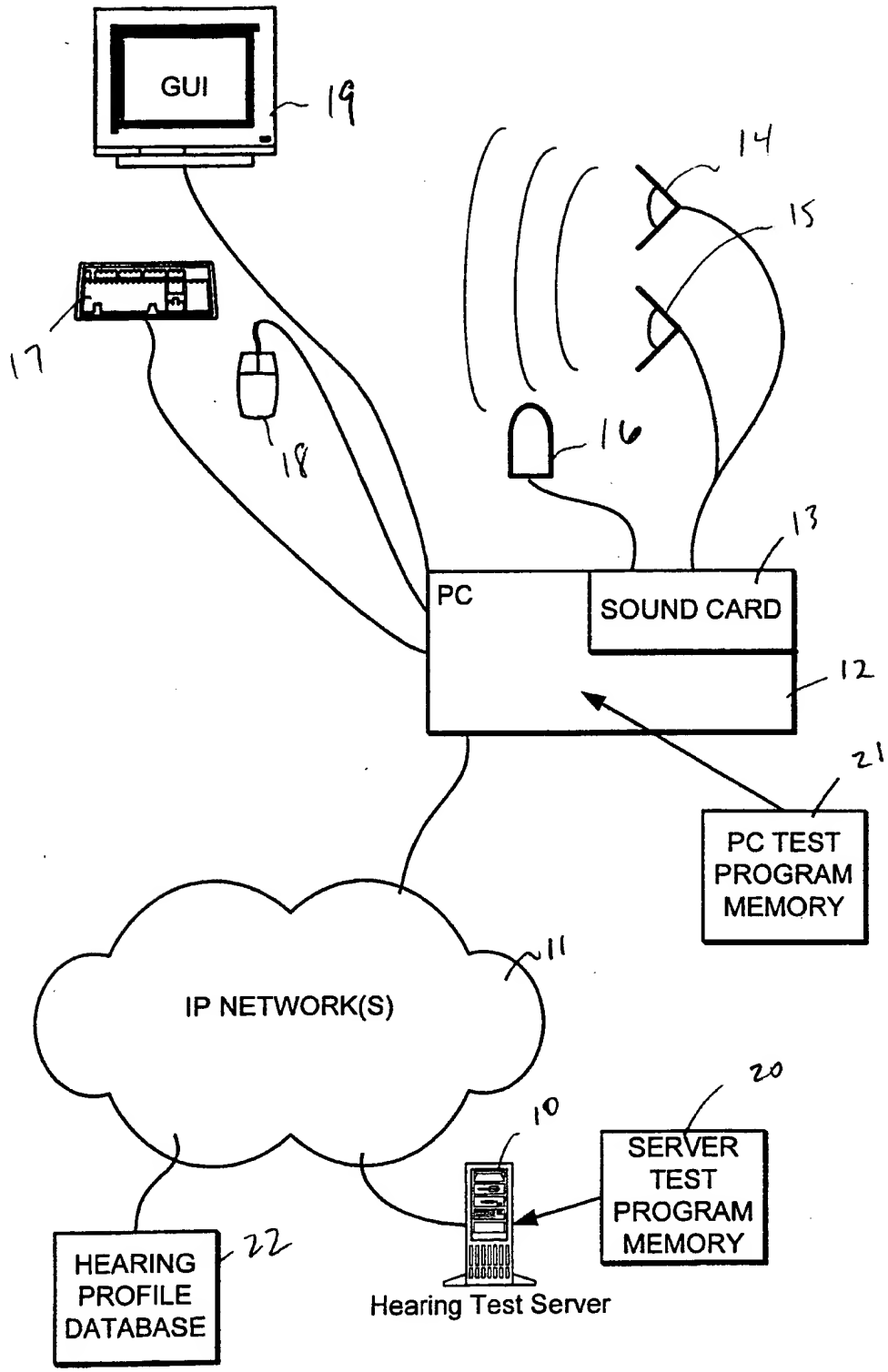
GUI
19
14
15
17
18
16
13
PC
SOUND CARD
12
21
PC TEST PROGRAM MEMORY
IP NETWORK(S)
11
20
10
SERVER TEST PROGRAM MEMORY
HEARING PROFILE DATABASE
22
Hearing Test Server

FIG. 1

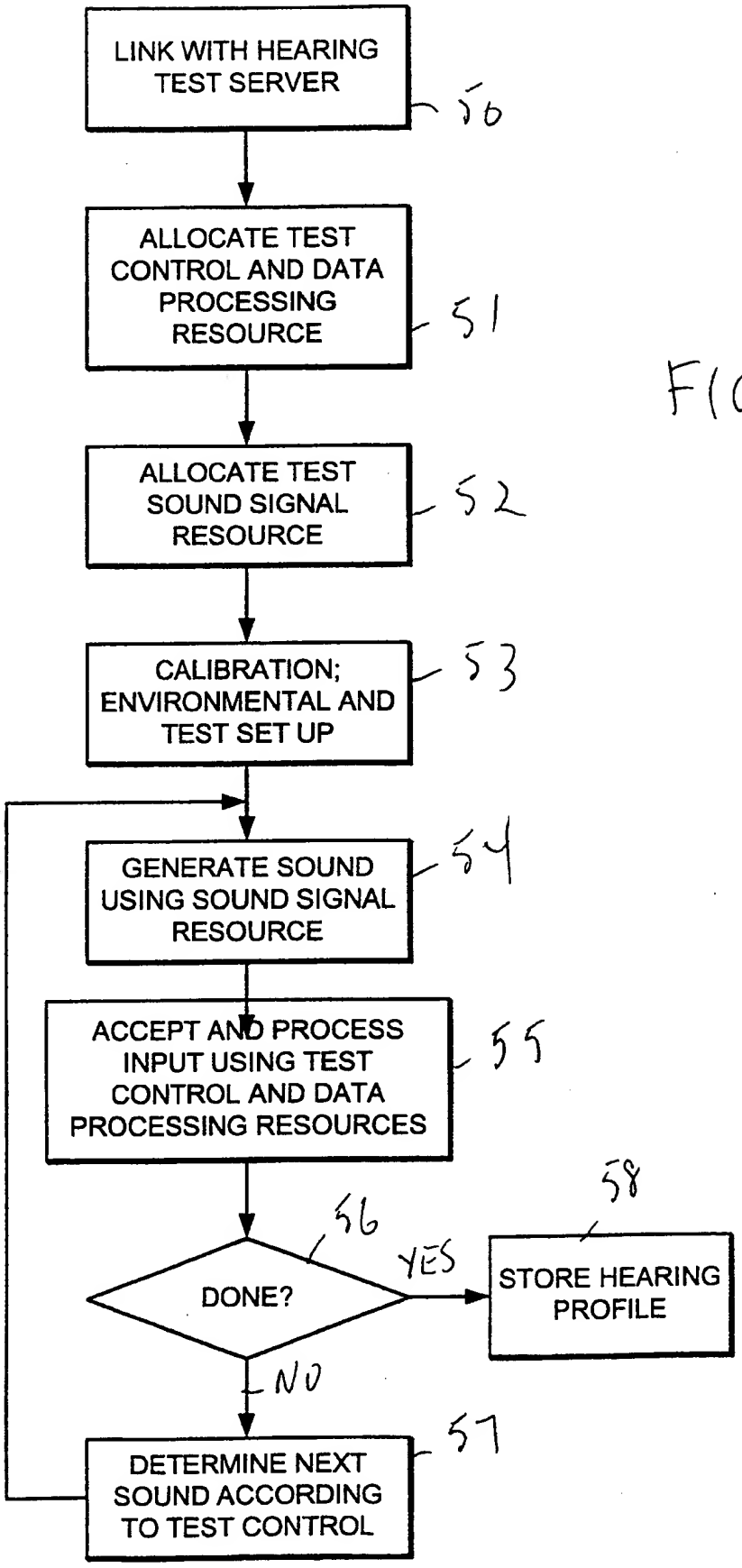
LINK WITH HEARING TEST SERVER
50
ALLOCATE TEST CONTROL AND DATA PROCESSING RESOURCE
51
ALLOCATE TEST SOUND SIGNAL RESOURCE
52
CALIBRATION; ENVIRONMENTAL AND TEST SET UP
53
GENERATE SOUND USING SOUND SIGNAL RESOURCE
54
ACCEPT AND PROCESS INPUT USING TEST CONTROL AND DATA PROCESSING RESOURCES
55
DONE?
56
YES
STORE HEARING PROFILE
58
NO
DETERMINE NEXT SOUND ACCORDING TO TEST CONTROL
57

FIG. 2

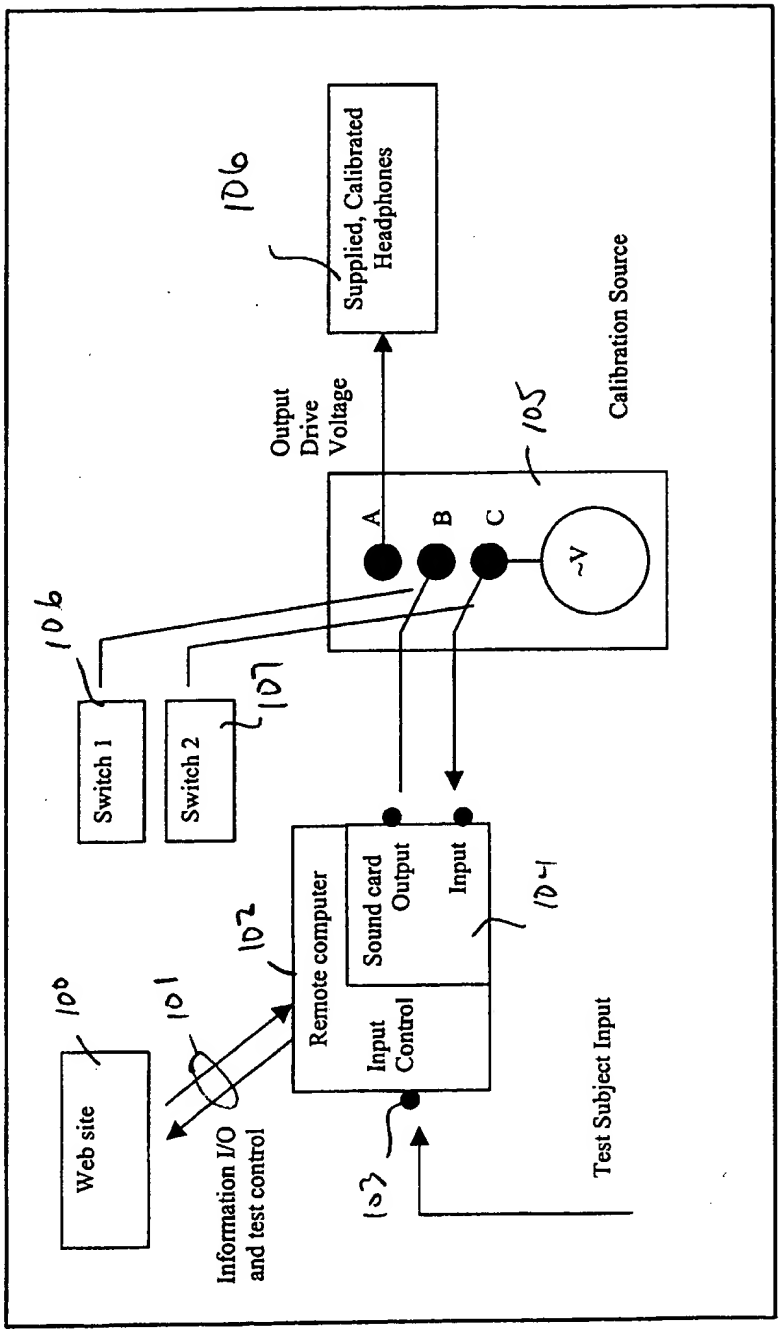
Supplied, Calibrated Headphones
106
Output Drive Voltage
105
Calibration Source
A
B
C
~V
106
Switch 1
Switch 2
107
Remote computer
Sound card Output
Input
104
Input Control
102
100
101
Web site
Information I/O and test control
103
Test Subject Input

FIG 3

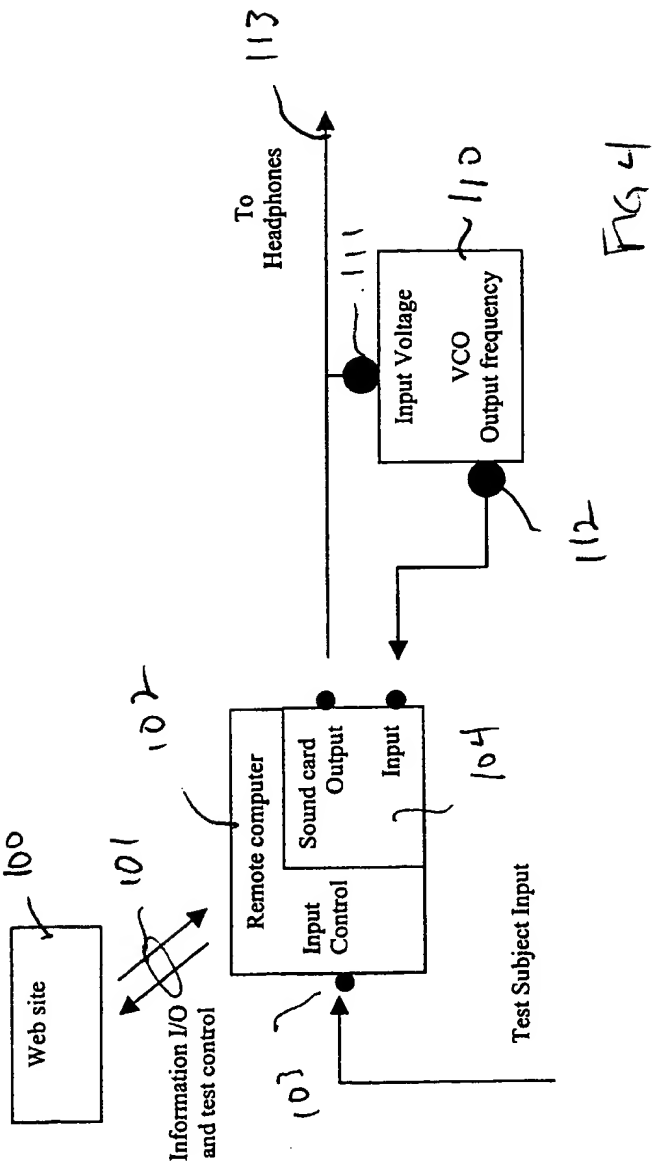
100
Web site
101
Information I/O and test control
102
Remote computer
Sound card Output
Input
Input Control
103
104
Test Subject Input
110
Input Voltage
VCO Output frequency
111
112
113
To Headphones

Fig 1

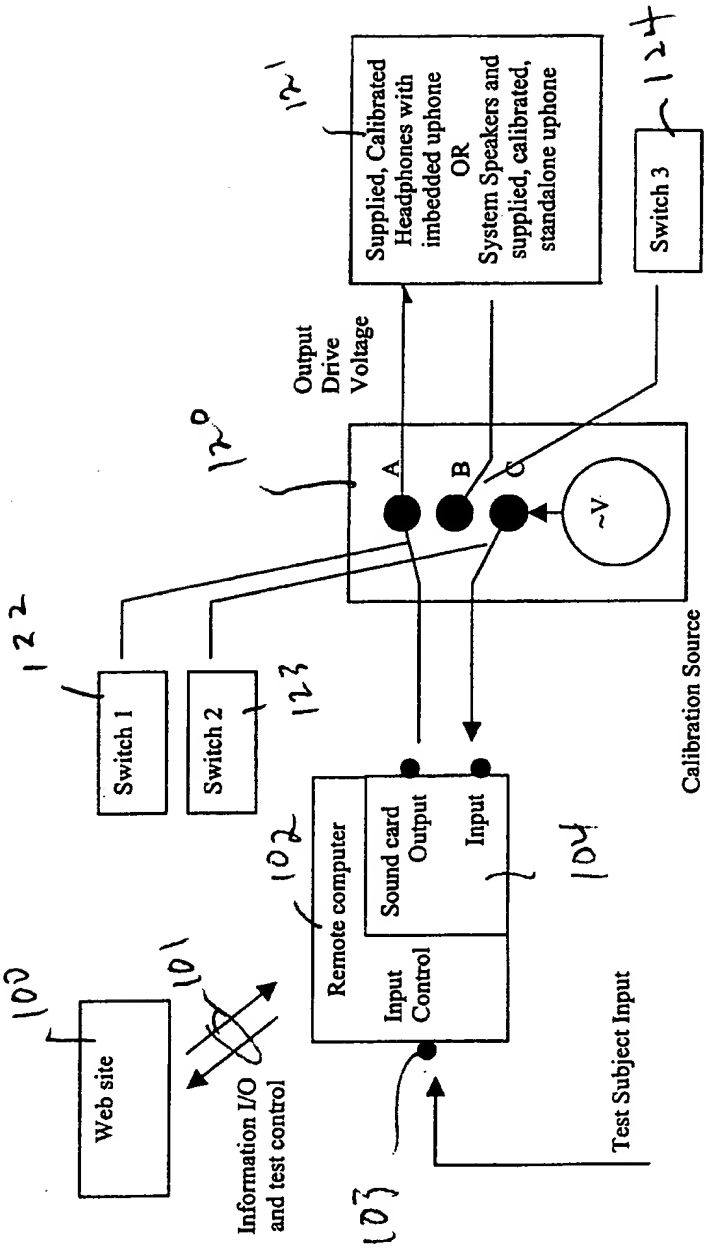
Supplied, Calibrated Headphones with imbedded uphone OR System Speakers and supplied, calibrated, standalone uphone
121
Switch 3
124
Output Drive Voltage
120
A
B
C
~V
122
Switch 1
Switch 2
123
Calibration Source
Remote computer
Sound card Output
Input
102
104
Input Control
Web site
100
101
Information I/O and test control
107
Test Subject Input

FIG. 5

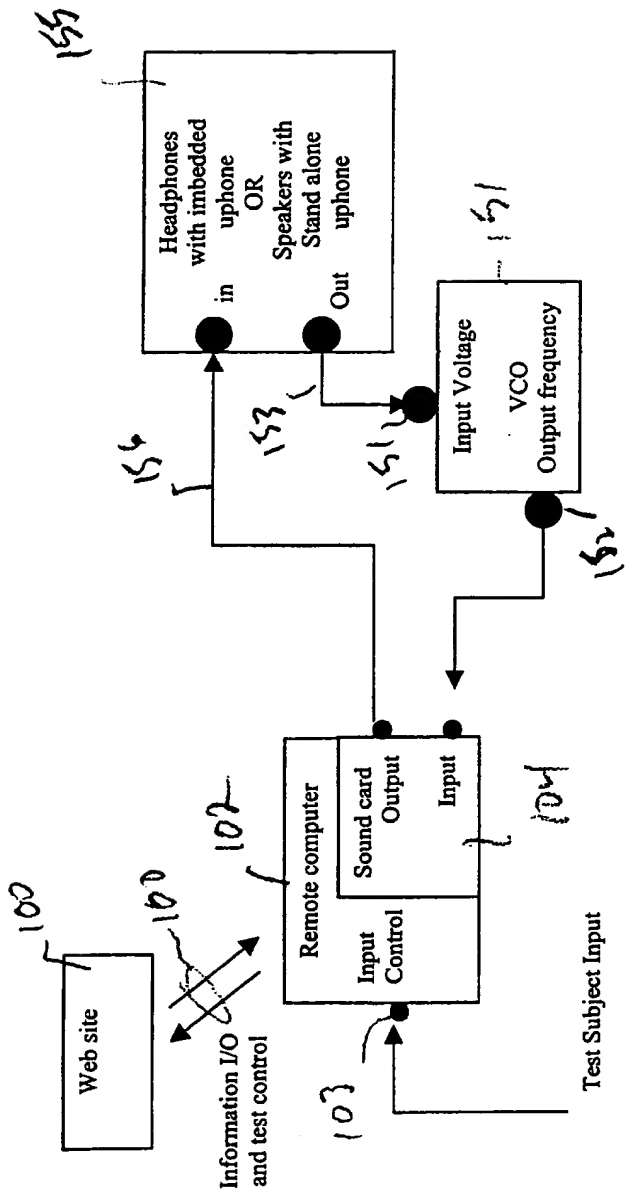
Headphones with imbedded uphone OR Speakers with Stand alone uphone
in
Out
Input Voltage
VCO
Output frequency
Remote computer
Sound card
Output
Input
Input Control
Web site
Information I/O and test control
Test Subject Input
155
151
153
156
152
102
104
100
101
103

FIG. 6

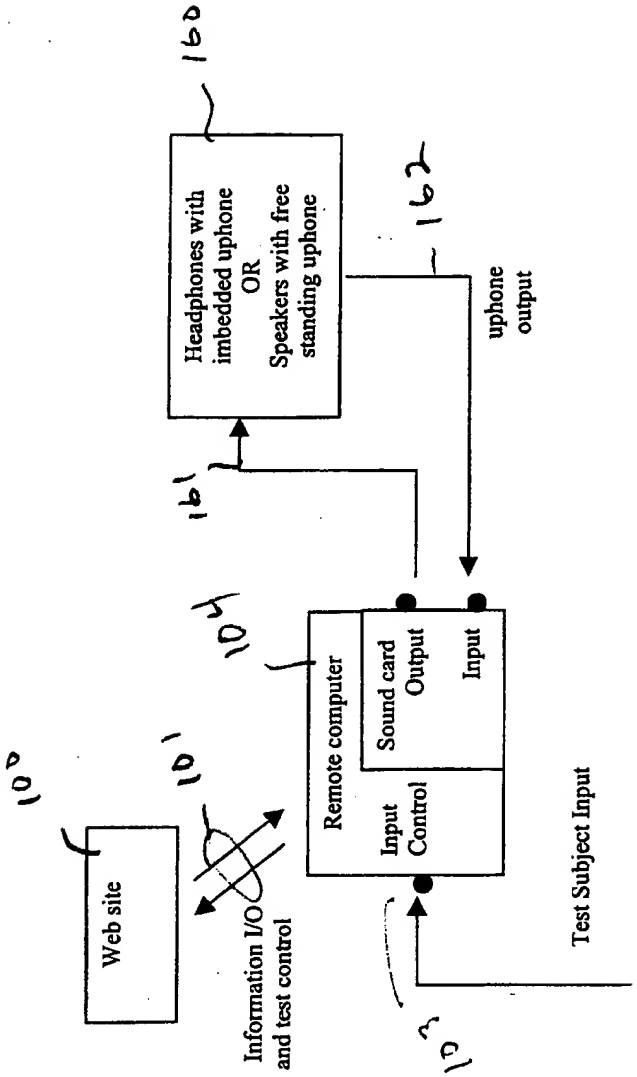
Web site
Information I/O and test control
Remote computer
Sound card Output
Input
Input Control
Test Subject Input
Headphones with imbedded uphone OR Speakers with free standing uphone
uphone output
10
10'
10''
12
16'
16''
16'''

FIG. 7

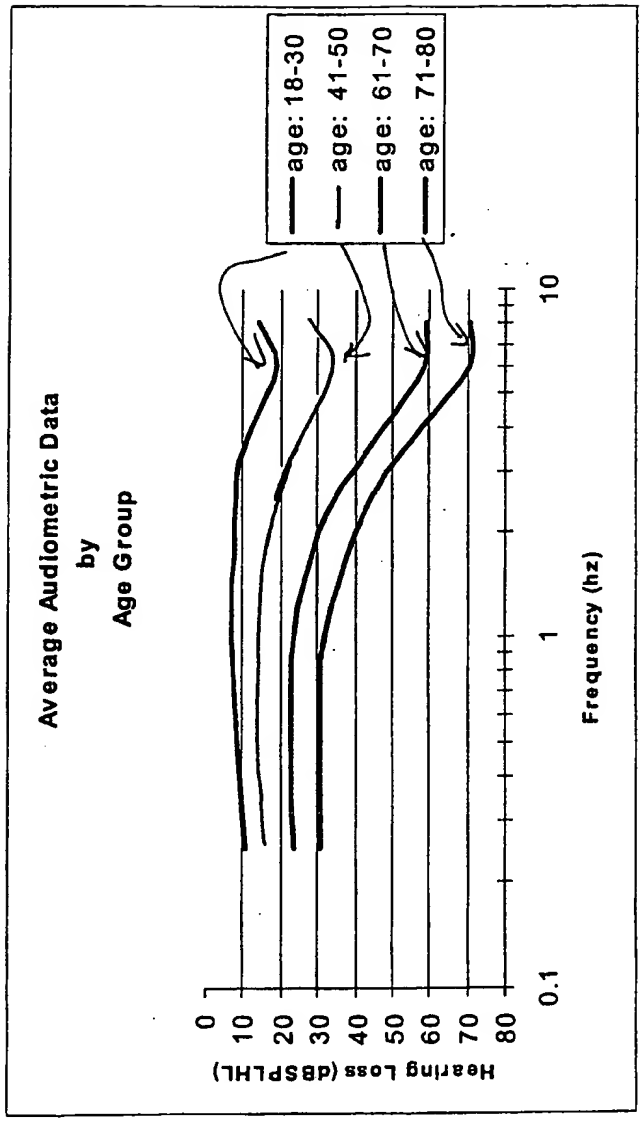
Average Audiometric Data
by
Age Group
age: 18-30
age: 41-50
age: 61-70
age: 71-80
Hearing Loss (dBSPLHL)
0
10
20
30
40
50
60
70
80
0.1
1
10
Frequency (hz)

FIG. 8